Claims

[c1] 1.A voice-over-Internet-Protocol (VOIP) application comprising:

a jitter buffer for receiving incoming VOIP packets from an Internet, the incoming VOIP packets containing compressed audio data from a remote VOIP application;

an audio application, receiving the compressed audio data from the incoming VOIP packets, for de-compressing the compressed audio data for playback to a user, and for capturing local audio data from the user;

a packetizer, receiving the local audio data from the audio application, for compressing the local audio data and encapsulating the local audio data into outgoing VOIP packets for transmission over the Internet to the remote VOIP application;

an estimator, receiving a reception time for a current VOIP packet that indicates when the current VOIP packet was received by the jitter buffer and receiving a prior reception time that indicates when a prior VOIP packet was received by the jitter buffer, for generating a bandwidth estimate for an incoming path through the Internet taken by the incoming VOIP packets; and

a comparator, in the estimator, for comparing an arrival difference of the reception time and the prior reception time to an audio duration that indicates a duration of playback to the user of the compressed audio data encapsulated by the current VOIP packet;

wherein the estimator decreases the bandwidth estimate when the comparator determines that the arrival difference exceeds the audio duration but increases the bandwidth estimate when the comparator determines that the arrival difference is below the audio duration;

wherein the packetizer receives the bandwidth estimate from the estimator, the packetizer sending the bandwidth estimate to the remote VOIP application, whereby incoming bandwidth is estimated by comparison of the arrival difference to the audio duration.

[c2] 2.The VOIP application of claim 1 wherein the estimator re-estimates the bandwidth estimate continuously for each incoming VOIP packet or periodically for a subset of the incoming VOIP packets.

[c3] 3.The VOIP application of claim 1 wherein the packetizer inserts the bandwidth estimate into the outgoing VOIP packets.

[c4] 4.The VOIP application of claim 1wherein the jitter buffer receives a send time for the current VOIP packet from the jitter buffer, the send time being added to the current VOIP packet by the remote VOIP application to indicate a time that the current VOIP packet was transmitted to the Internet by the remote VOIP application;

a congestion estimator, receiving the send time and the reception time for the current VOIP packet, for generating a current latency as a difference of the reception time and the send time for the current VOIP packet; and

a moving average comparator, receiving the current latency, for comparing the current latency to a moving-average latency;

wherein the congestion estimator increases a congestion estimate for the incoming path when the current latency exceeds the moving-average latency, but for decreasing the congestion estimate after the current latency falls below the moving-average latency;

wherein the packetizer receives the congestion estimate from the congestion estimator, the packetizer sending the congestion estimate to the remote VOIP application,

whereby congestion on the incoming path is estimated by comparison of the current latency to the moving-average latency.

[c5] 5.The VOIP application of claim 1 wherein the jitter buffer re-orders the incoming VOIP packets based on sequence numbers contained in the incoming VOIP packets,

whereby out-of-order incoming VOIP packets are re-ordered prior to audio playback.

[c6] 6.A computerized method for estimating conditions on a network path from a remote application to a local application comprising:

receiving incoming audio packets that include audio data from the remote application;

extracting a duration-time from a current packet that indicates a duration of

audio playing time of audio data contained in the current packet;
generating a receive-time for the current packet that indicates when the current packet was received by the local application;
calculating an inter-packet arrival time as a difference between receive-times for the current packet and a prior packet;
comparing the inter-packet arrival time to the duration-time of the current packet;
(1) when the inter-packet arrival time is greater than the duration-time, reducing a bandwidth estimate to indicate reduced available bandwidth of the network path;
(2) when the inter-packet arrival time is less than the duration-time, increasing the bandwidth estimate to indicate increased available bandwidth of the network path; and
including the bandwidth estimate for the current packet in an outgoing packet to the remote application,
whereby bandwidth estimates are made by the local application on audio packets received from the remote application and the bandwidth estimates are sent to the remote application.

[c7] 7.The computerized method of claim 6 wherein the bandwidth estimate is included in an audio packet sent from the local application to the remote application,
whereby the audio packet contains audio data from the local application but the bandwidth estimate for audio packets sent by the remote application.

[c8] 8.The computerized method of claim 7 further comprising:
(3) when the inter-packet arrival time is substantially equal to the duration-time, increasing the bandwidth estimate by a small fixed amount to test for an increased available bandwidth of the network path.

[c9] 9.The computerized method of claim 8 wherein reducing the bandwidth estimate comprises reducing the bandwidth estimate by a portion of a difference of the inter-packet arrival time and the duration-time;
wherein increasing the bandwidth estimate comprises increasing the bandwidth

estimate by a portion of a difference of the duration-time and the inter-packet arrival time,
whereby bandwidth estimate changes are in proportion to differences between the duration-time and the inter-packet arrival time.

[c10] 10.The computerized method of claim 9 wherein the portion is a multiple of the duration-time.

[c11] 11.The computerized method of claim 9 wherein the prior packet has a sequence number that is less than a sequence number for the current packet.

[c12] 12.The computerized method of claim 11 wherein the sequence number of the prior packet is one less than the sequence number of the current packet.

[c13] 13.The computerized method of claim 6 wherein the incoming audio packets include the current packet having a send-time indicating when the current packet was transmitted by the remote application;
further comprising:
generating a current latency for the current packet as a difference of the receive-time and the send-time for the current packet;
comparing the current latency to a moving average latency that is generated from previous audio packets received from the remote application;
(1) when the current latency is above the moving average latency, increasing a congestion estimate to indicate detection of congestion along the network path;
(2) when the current latency is substantially equal to the moving average latency, decreasing the congestion estimate to indicate network stability along the network path; and
including the congestion estimate for the current packet in the outgoing packet to the remote application,
whereby congestion estimates are made by the local application on audio packets received from the remote application and the congestion estimates are sent to the remote application.

[c14] 14.The computerized method of claim 13 further comprising:
(3) when the current latency is below the moving average latency, not changing

the congestion estimate to indicate recovery from congestion along the network path,
whereby congestion is allowed time to recover before the congestion estimate is reduced.

[c15] 15.The computerized method of claim 14 wherein the congestion estimate is increased in proportion to an absolute value of a difference between the current latency and the moving average latency but the congestion estimate is decreased by a fixed amount.

[c16] 16.The computerized method of claim 15 wherein the current latency is a one-way latency from the remote application to the local application and not a round-trip latency.

[c17] 17.A computer-program product comprising:
a computer-usable medium having computer-readable program code means embodied therein for estimating incoming bandwidth, the computer-readable program code means in the computer-program product comprising:
buffer means for receiving incoming packets sent by a remote audio application over a first network path;
wherein an incoming packet contains encoded remote audio data for replay to a local user and a duration value that indicate a duration of audio playback of the encoded remote audio data in the incoming packet;
audio means, receiving the encoded remote audio data, for decoding the encoded remote audio data for replay to the local user, and for encoding local audio captured from the local user to generated encoded local audio;
arrival timer means, coupled to the buffer means, for determining a delay between arrivals of the incoming packets;
analysis means for comparing the duration value to the delay from the arrival timer means and for adjusting a bandwidth estimate based on a comparison result; and
packeting means for generating outbound packets adding a packet header to segments of the encoded local audio, the packet header for assisting routing of the outbound packets to the remote audio application over a second network

path that can differ from the first network path;
wherein the packeting means sends the bandwidth estimate from the analysis means to the remote audio application, to indicate a condition of the first network path,
whereby current-status feedback of the first network path is sent to the remote audio application.

[c18] 18.The computer-program product of claim 17 wherein the packeting means inserts the bandwidth estimate from the analysis means into at least some of the outbound packets to provide current-status feedback to the remote audio application, the current-status feedback indicating a condition of the first network path,
whereby current-status feedback of the first network path is sent to the remote audio application with the encoded local audio.

[c19] 19.The computer-program product of claim 17 wherein the incoming packets also contain a send time-stamp;
further comprising:
congestion estimate means, coupled to receive the send time-stamp from the incoming packets, for determining a packet latency for routing of the incoming packets over the first network path but not over the second network path, the congestion estimate means generating a congestion estimate by comparing the packet latency to a moving average packet latency to signal increased congestion when the packet latency rises above the moving average packet latency,
wherein the packeting means sends the congestion estimate from the congestion estimate means to the remote audio application.

[c20] 20.The computer-program product of claim 17 further comprising:
packet loss means for increasing a packet loss counter when an incoming packet sent by the remote audio application fails to arrive at the buffer means within an acceptable delay,
wherein the packeting means sends the packet loss counter from the packet loss means to the remote audio application.